REMARKS

Claims 2-24 had been indicated to be allowable in the Office action mailed 02/03/2003. Accordingly, applicants had responded to that Office action by canceling claim 1 and rewriting in independent form those claims that had directly depended from claim 1. Applicants had not intended by that action to have acceded to the validity of the rejection of claim 1 set forth in the Office action of 02/03/2003. Indeed, applicants had intended to file a continuation application in which claim 1 would have been presented.

Claims 2-24 that were previously deemed to be directed to allowable subject matter were then rejected in the Office action of 09/02/2003. Applicants believe that the subject matter defined in all of the originally presented claims—including claim 1—distinguish the invention from the cited prior art. The Rules of Practice do not allow canceled claim 1 to be re-introduced with the same claim number. Rather than re-introduce a new claim similar to canceled claim 1 with a new claim number and then re-amend claims that had depended from claim 1, applicants have chosen to cancel claims 2-24 and to present new claims 25-76 which are directed, in part, to the inventive subject matter that had been defined in claims 2-24, albeit using somewhat different terminology.

Objections and Rejection under 35 USC 112

Various ones of claims 2-24 were objected to due to informalities or rejected as being indefinite under 35 USC 112. Since claims 2-24 have all been canceled, the objections and Section 112 rejection have been rendered moot. The examiner's careful reading of claims 2-24 is appreciated and her remarks vis-à-vis claims 2-24 were carefully taken into account when drafting new claims 25-76. It is believed that none of the newly added claims are informal or indefinite.

Rejections under 35 USC 102-103

Claims 2-24 were rejected as unpatentable over Sengodan or Sengodan in combination with Ash and/or Hin. Since these claims have all been canceled, those rejections have been rendered moot. However, it is noted for the record that applicants regard those claims, as well as previously canceled claim 1, as distinguishing the invention from the cited prior art, for at least some of the reasons set forth hereinbelow relative to new claims 25-76.

In the following sections, various aspects of applicants' invention as set forth in various ones of the claims are discussed. It is respectfully submitted that even if it would have been obvious to combine the teachings of the cited prior art references in some way or another, none of the cited references, taken singly or in combination, show or suggest these aspects of the invention and applicants' claims do not read on that which the references, or combinations of them, do disclose.

Reserving of Resources

Independent claims 25, 32, 68 and 74—like various ones of canceled claims 1-24—call for "reserving...packet network resources."

The Office action points to the operation in Sengodan, wherein an Admission Request message "carrying the bandwidth the endpoint requires for the duration of the call" as corresponding to the reserving function called for in applicants' claims.

Applicants respectfully disagree.

When resources are reserved, this means that they are guaranteed to be available for the packets to follow. Even when giving the wording "reserving" its broadest reasonable ordinary meaning, the function of "reserving resources" necessarily involves such a guarantee or a setting aside of the resources. The following definition of the verb "reserve" appears in *The Random House Dictionary of the English Language,* 1966 edition:

> **1**. to keep back or save for future use, disposal, treatment, etc. **2**. to retain or secure by express stipulation. **3**. to set apart for a particular use, purpose, service, etc.:...

By contrast, no actual reserving of resources is carried out in Sengodan. Rather, the ARQ message in Sengodan is an indication as to how much bandwidth the endpoint would like to have. It is not a reservation of resources and, indeed, there is nothing in Sengodan to suggest that a device that requested a certain amount of bandwidth is going to have resources reserved for it so as to guarantee that the call will actually be provided with that bandwidth throughout the call.

Indeed, as is well known, the H.323 protocol that is used in Sengodan is one that sits "on top of" common network architectures. See, for example, the H.323 primer (copy attached) appearing at http://www.packetizer.com/iptel/h323/papers/primer under the heading "Network Independence." This means that any allocation of resources on the network that is actually going to carry the call is not any concern of H.323 but, rather, of the underlying network.

Moreover, independent claims 25, 32, 68 and 74 all recite that what is being reserved are resources of two or more packet networks and that those two networks have different reservation policies. Sengodan does disclose the transmission of signals, e.g., packets, over different networks. However, since Sengodan does not disclose any resource reservation, that reference certainly cannot be said to disclose that two packet networks have different resource reservation policies.

Moreover, various claims set forth limitations as to the manner in which resources are reserved. Since as discussed above the prior art does not disclose the reserving of resources, it certainly cannot be said to disclose, for example,

a) reserving resources in particular kinds of networks, such as an access network, backbone network, television coaxial cable network, and packet telephony service provider network (e.g., claims **26, 27, 32, 33, 38, 40, 43, 45, 52, 54, 56, 58, 59, 62, 68, 69, and 73**);
b) indicating any kind of limits for reserved resources (e.g., claims **29 and 31**);
c) reserving resources on a per-call or multiple-call basis (e.g., claims **34, 35**

and 49);

d) reserving bi-directional or uni-directional capacity (e.g., claims **36, 37, 46, 47, and 66**);

e) reserving resources in a network based on a selected one of a plurality of reservation policies for that network (e.g., claims **44, 57, 70 and 75**);

f) reserving a constant-bit-rate channel in one network and other than a constant-bit-rate channel in another network (e.g., claims **61 and 72**).

Separate Reservation Policies for Interconnected Packet Networks

Each of the claims in the application makes reference to at least two networks, both of which are packet networks. Applicants are not aware of any prior art showing or suggesting that when a call is made using the resources of more than one packet network, that the networks could have different resource reservation policies and that the call could nonetheless be set up by what is essentially a two-step process in which resources for each packet network are reserved separately.

Certainly this idea is neither shown nor suggested in any of the cited prior art.

Reserving of Resources in Response to Reserve Messages

A further distinguishing aspect of applicants' invention relates to the fact that in particular embodiments of the invention, the resource reservation in, for example, an access packet network and a backbone packet network according to their own reservation policies is carried out in response to a single reserve message received from a calling or from a called party. See, for example, claims **38, 42, 44 and 55**. Even if the recitations in applicants' claims directed to reserving resources in two or more packet networks according to their own reservation policies could be said to read on prior art arrangements, the prior art certainly does not show or suggest doing any such reservation in response to a single reserve message received from a party.

A yet further distinguishing aspect of the invention relates to the fact that in particular embodiments of the invention, a second reserve message—specifically a backbone reserve message—is sent into the backbone packet network in response to the reserve message received from the calling party. See, for example, claims **39 and 44**. Nothing in the cited prior art shows or suggests this aspect of the invention.

And certainly, then, the prior art cannot be said to show or suggest the aspect of the invention claimed in, for example, claim **42**. That claim, together with its antecedent claims 36-38, recites that a) in response to a *calling* party's reserve message, resources are reserved in a *first* access packet network and a *backbone* network and that b) in response to a *called* party's reserve message, resources are reserved in a *second* access packet network and the backbone network, where the reservation policy for the backbone network is different from that of the access packet networks.

A Single Device Carries Out the Claimed Method

A further distinguishing aspect of applicants' invention relates to the fact that in particular embodiments of the invention, the resource reservation in at least two networks is carried out by a single device and, in particular embodiments, that particular device is a network edge device that may be a bridge or a router. See, for example, claims **51, 58, 60, 68, 71, 74 and 76**.

Other Distinguishing Aspects of the Invention

Various other limitations in the claims may provide yet further bases on which it can be found that applicants' claims distinguish over the prior art. Inasmuch as the foregoing is believed to establish the patentability of each of the claims in the application, such further points of distinction need not be raised at this time. Thus while

focusing herein on particular distinguishing aspects of the invention, Applicants may choose in the future to point to yet other bases on which various claims distinguish over the prior art.

In view of the foregoing, it is believed that all of the claims in the application are in condition for allowance. Reconsideration and passage of the application to issue are earnestly solicited.

Respectfully submitted,

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H.323

Packetizer™

A Primer on the H.323 Series Standard

Version 2.0

The H.323 standard provides a foundation for audio, video, and data communications across IP-based networks, including the Internet. By complying to H.323, multimedia products and applications from multiple vendors can interoperate, allowing users to communicate without concern for compatibility. H.323 will be the keystone for LAN-based products for consumer, business, entertainment, and professional applications.

H.323 is an umbrella recommendation from the International Telecommunications Union (ITU) that sets standards for multimedia communications over Local Area Networks (LANs) that do not provide a guaranteed Quality of Service (QoS). These networks dominate today's corporate desktops and include packet-switched TCP/IP and IPX over Ethernet, Fast Ethernet and Token Ring network technologies. Therefore, the H.323 standards are important building blocks for a broad new range of collaborative, LAN-based applications for multimedia communications.

The H.323 specification was approved in 1996 by the ITU's Study Group 16. Version 2 was approved in January 1998. The standard is broad in scope and includes both stand-alone devices and embedded personal computer technology as well as point-to-point and multipoint conferences. H.323 also addresses call control, multimedia management, and bandwidth management as well as interfaces between LANs and other networks.

H.323 is part of a larger series of communications standards that enable videoconferencing across a range of networks. Known as H.32X, this series includes H.320 and H.324, which address ISDN and PSTN communications, respectively. This primer provides an overview of the H.323 standard, its benefits, architecture, and applications.

Why H.323 is Important

The H.323 Recommendation is comprehensive, yet flexible, and can be applied to voice-only handsets and full multimedia video-conferencing stations, among others. H.323 applications are set to grow into the mainstream market for several reasons.

- H.323 sets multimedia standards for the existing infrastructure (i.e. IP-based networks). Designed to compensate for the effect of highly variable LAN latency, H.323 allows customers to use multimedia applications without changing their network infrastructure.
- IP LANs are becoming more powerful. Ethernet bandwidth is migrating from 10 Mbps to 100 Mbps, and Gigabit Ethernet is making headway into the market.
- By providing device-to-device, application-to-application, and vendor-to-vendor interoperability, H.323 allows customer products to interoperate with other H.323-compliant products.
- PCs are becoming more powerful multimedia platforms due to faster processors, enhanced instruction sets, and powerful multimedia accelerator chips.

- H.323 provides standards for interoperability between LANs and other networks.
- Network loading can be managed. With H.323, the network manager can restrict the amount of network bandwidth available for conferencing. Multicast support also reduces bandwidth requirements.
- H.323 has the support of many computing and communications companies and organizations, including Intel, Microsoft, Cisco, and IBM. The efforts of these companies will generate a higher level of awareness in the market.

Key Benefits of H.323

Codec Standards

H.323 establishes standards for compression and decompression of audio and video data streams, ensuring that equipment from different vendors will have some area of common support.

Interoperability

Users want to conference without worrying about compatibility at the receiving point. Besides ensuring that the receiver can decompress the information, H.323 establishes methods for receiving clients to communicate capabilities to the sender. The standard also establishes common call setup and control protocols.

Network Independence

H.323 is designed to run on top of common network architectures. As network technology evolves, and as bandwidth-management techniques improve, H.323-based solutions will be able to take advantage of those enhanced capabilities.

Platform and Application Independence

H.323 is not tied to any hardware or operating system. H.323-compliant platforms will be available in many sizes and shapes, including video-enabled personal computers, dedicated platforms, IP-enabled telephone handsets, cable TV set-top boxes and turnkey boxes.

Multipoint Support

Although H.323 can support conferences of three or more endpoints without requiring a specialized multipoint control unit, MCU's provide a more powerful and flexible architecture for hosting multipoint conferences. Multipoint capabilities can be included in other components of an H.323 system.

Bandwidth Management

Video and audio traffic is bandwidth-intensive and could clog the corporate network. H.323 addresses this issue by providing bandwidth management. Network managers can limit the number of simultaneous H.323 connections within their network or the amount of bandwidth available to H.323 applications. These limits ensure that critical traffic will not be disrupted.

Multicast Support

H.323 supports multicast transport in multipoint conferences. Multicast sends a single packet to a subset of destinations on the network without replication. In contrast, unicast sends multiple point-to-point transmissions, while broadcast sends to all destinations. In unicast or broadcast, the network is used inefficiently as packets are replicated throughout the network. Multicast transmission uses bandwidth more efficiently since all stations in the multicast group read a single data stream.

Flexibility

An H.323 conference can include endpoints with different capabilities. For example, a terminal with audio-only capabilities can participate in a conference with terminals that have video and/or data capabilities. Furthermore, an H.323 multimedia terminal can share the data portion of a video conference with a T.120 data-only terminal, while sharing voice, video, and data with other H.323 terminals.

Inter-Network Conferencing

Many users want to conference from a LAN to a remote site. For example, H.323 establishes a means of linking LAN-based desktop systems with ISDN-based group systems. H.323 uses common codec technology from different videoconferencing standards to minimize transcoding delays and to provide optimum performance.

Architectural Overview

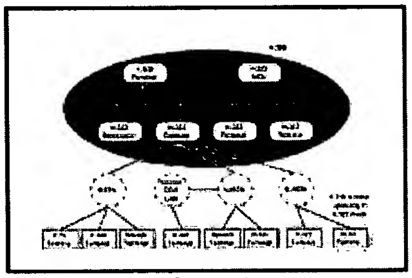

figure 1
Click for larger version

The H.323 Recommendation covers the technical requirements for audio and video communications services in LANs that do not provide a guaranteed Quality of Service (QoS). H.323 references the T.120 specification for data conferencing and enables conferences which include a data capability. The scope of H.323 does not include the LAN itself or the transport layer that may be used to connect various LANs. Only elements needed for interaction with the Switched Circuit Network (SCN) are within the scope of H.323. Figure 1 outlines an H.323 system and its components.

H.323 defines four major components for a network-based communications system: Terminals, Gateways, Gatekeepers, and Multipoint Control Units.

Terminals

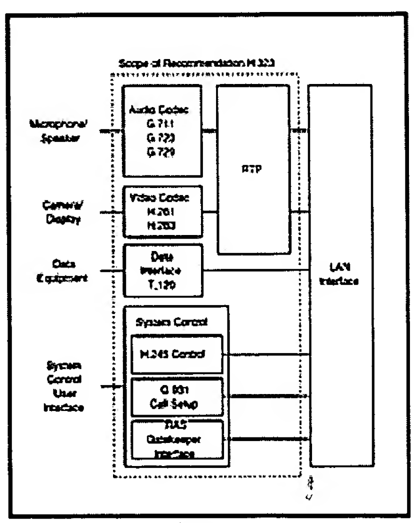

figure 2
Click for larger version

Terminals are the client endpoints on the LAN that provide real-time, two-way communications. Figure

2 describes the terminal components. All terminals must support voice communications; video and data are optional. H.323 specifies the modes of operation required for different audio, video, and/or data terminals to work together. It is the dominant standard of the next generation of Internet phones, audio conferencing terminals, and video conferencing technologies.

All H.323 terminals must also support H.245, which is used to negotiate channel usage and capabilities. Three other components are required: Q.931 for call signaling and call setup, a component called Registration/Admission/Status (RAS), which is a protocol used to communicate with a Gatekeeper; and support for RTP/RTCP for sequencing audio and video packets.

Optional components in an H.323 terminal are video codecs, T.120 data conferencing protocols, and MCU capabilities (described further below).

Gateways

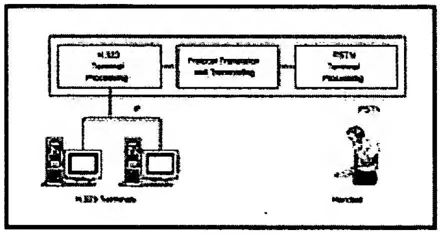

figure 3
Click for larger version

The Gateway is an optional element in an H.323 conference. Gateways provide many services, the most common being a translation function between H.323 conferencing endpoints and other terminal types. This function includes translation between transmission formats (i.e. H.225.0 to H.221) and between communications procedures (i.e. H.245 to H.242). In addition, the Gateway also translates between audio and video codecs and performs call setup and clearing on both the LAN side and the switched-circuit network side. Figure 3 shows an H.323/PSTN Gateway.

In general, the purpose of the Gateway is to reflect the characteristics of a LAN endpoint to an SCN endpoint and vice versa. The primary applications of Gateways are likely to be:

- Establishing links with analog PSTN terminals.
- Establishing links with remote H.320-compliant terminals over ISDN-based switched-circuit networks.
- Establishing links with remote H.324-compliant terminals over PSTN networks

Gateways are not required if connections to other networks are not needed, since endpoints may directly communicate with other endpoints on the same LAN. Terminals communicate with Gateways using the H.245 and Q.931 protocols.

With the appropriate transcoders, H.323 Gateways may support terminals that comply with H.310, H.321, H.322, and V.70.

Many Gateway functions are left to the designer. For example, the actual number of H.323 terminals that can communicate through the Gateway is not subject to standardization. Similarly, the number of SCN connections, the number of simultaneous independent conferences supported, the audio/video/data conversion functions, and inclusion of multipoint functions are left to the manufacturer. By incorporating Gateway technology into the H.323 specification, the ITU has positioned H.323 as the glue that holds the world of standards-based conferencing endpoints together.

Gatekeepers

A Gatekeeper is the most important component of an H.323 enabled network. It acts as the central point for all calls within its zone and provides call control services to registered endpoints. In many ways, an

H.323 gatekeeper acts as a virtual switch.

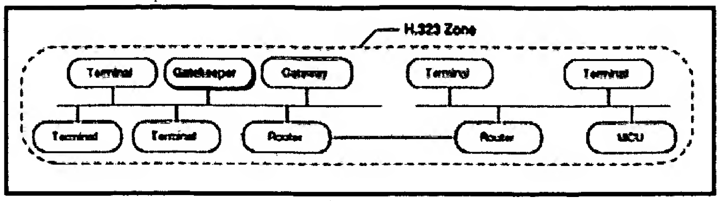

figure 4
Click for larger version

Gatekeepers perform two important call control functions. The first is address translation from LAN aliases for terminals and gateways to IP or IPX addresses, as defined in the RAS specification. The second function is bandwidth management, which is also designated within RAS. For instance, if a network manager has specified a threshold for the number of simultaneous conferences on the LAN, the Gatekeeper can refuse to make any more connections once the threshold is reached. The effect is to limit the total conferencing bandwidth to some fraction of the total available; the remaining capacity is left for e-mail, file transfers, and other LAN protocols. The collection of all Terminals, Gateways, and Multipoint Control Units managed by a single gatekeeper is known as an H.323 Zone (Figure 4).

An optional, but valuable feature of a gatekeeper is its ability to route H.323 calls. By routing a call through a gatekeeper, it can be controlled more effectively. Service providers need this ability in order to bill for calls placed through their network. This service can also be used to re-route a call to another endpoint if a called endpoint is unavailable. In addition, a gatekeeper capable of routing H.323 calls can help make decisions involving balancing among multiple gateways. For instance, if a call is routed through a gatekeeper, that gatekeeper can then re-route the call to one of many gateways based on some proprietary routing logic.

While a Gatekeeper is logically separate from H.323 endpoints, vendors may incorporate Gatekeeper functionality into the physical implementation of Gateways and MCUs.

A Gatekeeper is not required in an H.323 system. However, if a Gatekeeper is present, terminals must make use of the services offered by gatekeepers. RAS defines these as address translation, admissions control, bandwidth control, and zone management.

Gatekeepers can also play a role in multipoint connections. To support multipoint conferences, users would employ a Gatekeeper to receive H.245 Control Channels from two terminals in a point-to-point conference. When the conference switches to multipoint, the Gatekeeper can redirect the H.245 Control Channel to a multipoint controller, the MC. The Gatekeeper need not process the H.245 signaling; it only needs to pass it between the terminals or the terminals and the MC.

LANs which contain Gateways could also contain a Gatekeeper to translate incoming E.164 addresses into Transport Addresses. Because a Zone is defined by its Gatekeeper, H.323 entities that contain an internal Gatekeeper require a mechanism to disable the internal function so that when there are multiple H.323 entities that contain a Gatekeeper on a LAN, the entities can be configured into the same Zone..

Required Gatekeeper Functions

Address Translation	Translation of alias address to Transport Address using a table that is updated with Registration messages. Other methods of updating the translation table are also allowed.
Admissions Control	Authorization of LAN access using Admission Request, Confirm and Reject (ARQ/ARC/ARJ) messages. LAN access may be based on call authorization, bandwidth, or some other criteria. Admissions Control may also be a null function which admits all requests.
Bandwidth	Support for Bandwidth Request, Confirm and Reject

Control	(BRQ/BCF/BRJ) messages. This may be based on bandwidth management. Bandwidth Control may also be a null function which accepts all requests for bandwidth changes.
Zone Management	The Gatekeeper provides the above functions for terminals, MCUs, and Gateways which have registered within its Zone of control.

Optional Gatekeeper Functions Include:

Call Control Signaling	In a point to point conference, the Gatekeeper may process Q.931 call control signals. Alternatively, the Gatekeeper may send the endpoints G.931 signals directly to each other.
Call Authorization	The Gatekeeper may reject a call from a terminal based on the Q.931 specification. The reasons for rejection may include, but are not limited to, restricted access to/from particular terminals or Gateways, restricted access during certain periods of time. The criteria for determining if authorization passes or fails is outside the scope of H.323.
Bandwidth Management	The Gatekeeper may reject calls from a terminal if it determines that sufficient bandwidth is not available. This function also operates during an active call if a terminal requests additional bandwidth. The criteria for determining if bandwidth is available is outside the scope of H.323.
Call Management	The Gatekeeper may maintain a list of ongoing H.323 calls in order to indicate that a called terminal is busy or to provide information for the Bandwidth Management function.

Multipoint Control Units (MCU)

The Multipoint Control Unit (MCU) supports conferences between three or more endpoints. Under H.323, an MCU consists of a Multipoint Controller (MC), which is required, and zero or more Multipoint Processors (MP). The MC handles H.245 negotiations between all terminals to determine common capabilities for audio and video processing. The MC also controls conference resources by determining which, if any, of the audio and video streams will be multicast.

The MC does not deal directly with any of the media streams. This is left to the MP, which mixes, switches, and processes audio, video, and/or data bits. MC and MP capabilities can exist in a dedicated component or be part of other H.323 components.

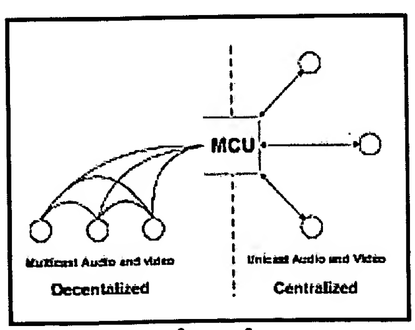

figure 5
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Multipoint Conferences

Multipoint conference capabilities are handled in a variety of methods and configurations under H.323. The Recommendation uses the concepts of centralized and decentralized conferences, as described in Figure 5.

Centralized multipoint conferences require the existence of an MCU to facilitate a multipoint conference. All terminals send audio, video, data, and control streams to the MCU in a point-to-point fashion. The MC centrally manages the conference using H.245 control functions that also define the capabilities for each terminal. The MP does the audio mixing, data distribution, and video switching/ mixing functions typically performed in multipoint conferences and sends the resulting streams back to the participating terminals. The MP may also provide conversion between different codecs and bit rates and may use multicast to distribute processed video. A typical MCU that supports centralized multipoint conferences consists of an MC and an audio, video, and/or data MP.

Decentralized multipoint conferences can make use of multicast technology. Participating H.323 terminals multicast audio and video to other participating terminals without sending the data to an MCU. Note that control of multipoint data is still centrally processed by the MCU, and H.245 Control Channel information is still transmitted in a point-to-point mode to an MC.

Receiving terminals are responsible for processing the multiple incoming audio and video streams. Terminals use H.245 Control Channels to indicate to an MC how many simultaneous video and audio streams they can decode. The number of simultaneous capabilities of one terminal does not limit the number of video or audio streams which are multicast in a conference. The MP can also provide video selection and audio mixing in a decentralized multipoint conference.

Hybrid multipoint conferences use a combination of centralized and decentralized features. H.245 signals and either an audio or video stream is processed through point-to-point messages to the MCU. The remaining signal (audio or video) is transmitted to participating H.323 terminals through multicast.

One advantage of centralized conferencing is that all H.323 terminals support point-to-point communications. The MCU may output multiple unicasts to the conference participants and no special network capabilities are required. Alternatively, the MCU may receive multiple unicasts, mix audio and switch video, and output a multicast stream, conserving network bandwidth.

H.323 also supports mixed multipoint conferences in which some terminals are in a centralized conference, others are in a decentralized conference, and an MCU provides the bridge between the two types. The terminal is not aware of the mixed nature of the conference, only of the mode of conference in which it sends and receives.

By supporting both multicast and unicast approaches, H.323 spans current generation and future networking technologies. Multicast makes more efficient use of network bandwidth, but places higher computational loads on the terminals, which have to mix and switch their own audio/video receiving streams. Additionally, multicast support is required in network routers and switches.

An MC may be located within a Gatekeeper, Gateway, Terminal, or MCU.

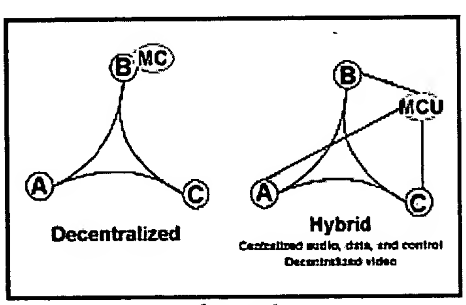

figure 6
Click for larger version

Consider a simple example where a multipoint conference is set up between three clients (Figure 6). One

client terminal (Client B) performs the MC function. All the terminals could use multicast to participate in a decentralized conference. An MP function on each node would mix and present the incoming audio and video signals to the user. This approach minimizes the need for specialized network resources. However, the network must be configured to support multicast.

A separate MCU can be used to handle only the audio, data, and control functions. In this configuration the video may still be multicast, which conserves bandwidth. This MCU could be either a dedicated system or a terminal with extra horsepower.

Multipoint conferences that span terminals on the LAN and off-network are likely to benefit from configurations where the MCU functions are tightly integrated with the Gateway.

H.323 Version 2

Approved in January of 1998, version 2 of the H.323 standard addresses deficiencies in version 1 and introduces new functionality within existing protocols, such as Q.931, H.245 and H.225, as well as entirely new protocols. The most significant advances were in security, fast call setup, supplementary services and T.120/H.323 integration.

Security

In development for months, the H.235 standard addresses four general issues when dealing with security, Authentication, Integrity, Privacy, and non-Repudiation. Authentication is a mechanism to make sure that the endpoints participating in the conference are really who they say they are. Integrity provides a means to validate that the data within a packet is indeed an unchanged representation of the data. Privacy/Confidentiality is provided by encryption and decryption mechanisms that hide the data from eavesdroppers so that if it is intercepted, it cannot be viewed. Non-Repudiation is a means of protection against someone denying that they participated in a conference when you know they were there.

Fast Call Setup

Using version one of H.323, a call was placed from one endpoint to another, but streams were not immediately available. This resulted in a long delay between the time a call was answered and when the participants could hear each other. With H.323 version two and the introduction of Fast Call Setup, this problem has been eliminated.

Supplementary Services

Supplementary Services for H.323, namely Call Transfer and Call Diversion, have been defined by the H.450 series. H.450.1 defines the signaling protocol between H.323 endpoints for the control of supplementary services. H.450.2 defines Call Transfer and H.450.3 Call Diversion. Call Transfer allows a call established between endpoint A and endpoint B to be transformed into a new call between endpoint B and a third endpoint, endpoint C. Call Diversion provides the supplementary services Call Forwarding Unconditional, Call Forwarding Busy, Call Forwarding No Reply and Call Deflection.

T.120/H.323 Integration

Although the first version of H.323 addressed the integration of T.120 with H.323, the call setup scenarios were somewhat complex and unclear. Version 2 of H.323 addresses this problem by requiring endpoints that support both T.120 and H.323 to lead the call with H.323. Further, version 2 states that T.120 is an optional part of an H.323 conference and that enabling T.120 is at the discretion of each H.323 endpoint.

Communications Under H.323

Communications under H.323 can be considered a mix of audio, video, data, and control signals. Audio capabilities, Q.931 call setup, RAS control, and H.245 signaling are required. All other capabilities, including video and data conferencing are optional. When multiple algorithms are possible, the algorithms used by the encoder are derived from information passed by the decoder during the H.245 capability exchange. H.323 terminals are also capable of asymmetric operation (different encode and decode algorithms) and can send/receive more than one video and audio channel.

Control

The call control functions are the heart of the H.323 terminal. These functions include signaling for call setup, capability exchange, signaling of commands and indications, and messages to open and describe the content of logical channels. All audio, video, and control signals pass through a control layer that formats the data streams into messages for output to the network interface. The reverse process takes place for incoming streams. This layer also performs logical framing, sequence numbering, error detection, and error correction as appropriate to each media type. The Q.931, RAS, and RTP/RTCP protocols perform these functions.

Overall system control is provided by three separate signaling functions: the H.245 Control Channel, the Q.931 Call Signalling Channel, and the RAS Channel.

The H.245 Control Channel is a reliable channel that carries control messages governing operation of the H.323 entity, including capabilities exchange, opening and closing of logical channels, preference requests, flow control messages, and general commands and indications. Capabilities exchange is one of the fundamental capabilities in the ITU recommendation; H.245 provides for separate receive and transmit capabilities as well as for methods to describe these details to other H.323 terminals. There is only one H.245 Control Channel per call.

The Call Signalling Channel uses Q.931 to establish a connection between two terminals.

The RAS signaling function performs registration, admission, bandwidth changes, status, and disengage procedures between endpoints and Gatekeepers. RAS is not used if a Gatekeeper is not present.

Audio

Audio signals contain digitized and compressed speech. The compression algorithms supported by H.323 are all proven ITU standards. H.323 terminals must support the G.711 voice standard for speech compression. Support for other ITU voice standards is optional.

The different ITU recommendations for digitizing and compressing speech signals reflect different tradeoffs between speech quality, bit rate, computer power, and signal delay. G.711 generally transmits voice at 56 or 64 kbps, well within the bandwidth limits likely on a LAN, but was designed originally for continuous bit-rate networks. Because G.723 operates at very low bit rates, it is strongly being considered as a required codec and will be the predominant audio codec in H.323 applications.

Video

While video capabilities are optional, any video-enabled H.323 terminal must support the H.261 codec; support for H.263 is optional. Video information is transmitted at a rate no greater than that selected during the capability exchange. H.261, which provides a measure of compatibility across many of the different ITU recommendations (see table on next page), is used with communication channels that are multiples of 64 kbps (P=1,2,3...30). H.261 calls for fully encoding some frames and for coding only the

difference between a frame and the previous frame in other cases. Motion compensation, which improves image quality, is an H.261 option.

H.263 is a backwards-compatible update to H.261. H.263 picture quality is greatly improved by using a required 1/2 pixel motion-estimation technique, predicted frames, and a Huffman coding table optimized for low bit rate transmissions. H.263 defines five standardized picture formats. Communications between H.261 systems and H.263 systems is facilitated because both must support QCIF.

Videoconferencing Picture Format	Image Size In Pixels	H.261	H.263
sub-QCIF	128x96	not specified	required
QCIF	176x144	required	required
CIF	352x288	optional	optional
4CIF	704x576	N/A	optional
16CIF	1408x1152	N/A	optional

ITU Formats for videoconferencing

Data

Data conferencing is an optional capability. When supported, data conferencing enables collaboration through applications such as shared whiteboards, application sharing, and file transfer.

H.323 supports data conferencing through the T.120 specification (Figure 7). An ITU standard, T.120 addresses point-to-point and multipoint data conferences. It provides interoperability at the application, network, and transport level.

An H.323 system can support data by incorporating T.120 capabilities into clients and multipoint control units. The MCU may control and mix the data conferencing information.

A recommendation for multicast support in T.120, known as T.125 Annex A or the Multicast Adaptation Protocol, was approved by the ITU in January of 1998

(A primer on the T.120 series standard is available.)

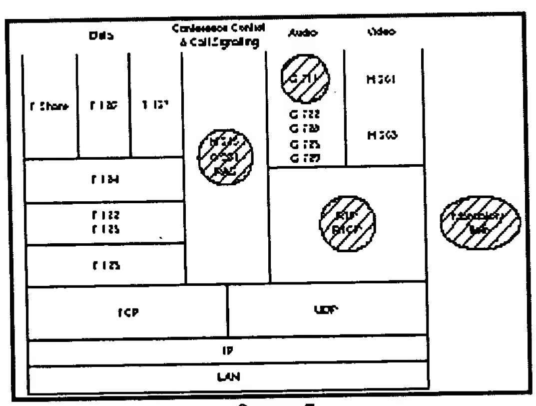

figure 7
Click for larger version

IP Networking and Multimedia Conferencing

H.323 uses both reliable and unreliable communications. Control signals and data require reliable transport because the signals must be received in the order in which they were sent and cannot be lost. Audio and video streams lose their value with time. If a packet is delayed, it may not have relevance to the end user. Audio and video signals use the more efficient unreliable transport.

Reliable transmission of messages uses a connection-oriented mode for data transmission. In the IP stack, this type of transmission is accomplished with TCP. Reliable transmission guarantees sequenced, error-free, flow-controlled transmission of packets, but can delay transmission and reduce throughput. H.323 uses reliable (TCP) end-to-end services for the H.245 Control Channel, the T.120 Data Channels, and the Call Signaling Channel.

Within the IP stack, unreliable services are provided by User Datagram Protocol (UDP). Unreliable transmission is a mode without connections that promises nothing more than "best effort" delivery. UDP offers minimal control information. H.323 uses UDP for the audio, video, and the RAS Channel.

In conferences with multiple audio and video streams, unreliable transport via UDP uses IP Multicast and the Real-Time Protocol (RTP) developed by the Internet Engineering Task Force (IETF) to handle streaming audio and video. IP Multicast is a protocol for unreliable multicast transmission in UDP. RTP works on top of IP Multicast, and was designed to handle the requirements of streaming audio and video over the Internet. A header containing a time-stamp and a sequence number is added to each UDP packet. With appropriate buffering at the receiving station, timing and sequence information allows the application to eliminate duplicate packets; reorder out-of-sequence packets; synchronize sound, video and data; and achieve continuous playback in spite of varying latencies.

Because H.323 is RTP-based, it can operate on the Internet's Multicast Backbone (Mbone), a virtual network on top of the Internet that provides a multicast facility and supports video, voice, and data conferencing. The Real-Time Control Protocol (RTCP) is used for the control of RTP. RTCP monitors the quality of service, conveys information about the session participants, and periodically distributes control packets containing quality information to all session participants through the same distribution mechanisms as the data packets.

Having sufficient bandwidth for a multimedia call is critical and difficult to ensure in large packet networks like the Internet or a corporate intranet. Another IETF protocol, the Resource Reservation Protocol (RSVP), allows a receiver to request a specific amount of bandwidth for a particular data stream and receive a reply indicating whether the request has been granted. Although RSVP is not an official part of the H.323 standard, some H.323 products will support it. RTP needs to be supported by Terminals, Gateways, and MCUs with Multipoint Processors. RSVP may also be supported by the same components and any intermediate switches or routers.

	H.320	**H.321**	**H.322**	**H.323 V1/V2**	**H.324**
Approval Date	1990	1995	1995	1996/1998	1996
Network	Narrowband switched digital ISDN	Broadband ISDN ATM LAN	Guaranteed bandwidth packet switched networks	Non-guaranteed bandwidth packet switched networks, (Ethernet)	PSTN or POTS, the analog phone system
Video	H.261 H.263	H.261 H.263	H.261 H.263	H.261 H.263	H.261 H.263
Audio	G.711 G.722 G.728	G.711 G.722 G.728	G.711 G.722 G.728	G.711 G.722 G.728 G.723 G.729	G.723
Multiplexing	H.221	H.221	H.221	H.225.0	H.223
Control	H.230 H.242	H.242	H.242 H.230	H.245	H.245
Multipoint	H.231 H.243	H.231 H.243	H.231 H.243	H.323	

Data	T.120	T.120	T.120	T.120	T.120
Comm. Interface	I.400	AAL I.363 AJM I.361 PHY I.400	I.400& TCP/IP	TCP/IP	V.34 Modem

Overview of ITU Videoconferencing Standards

H.323 is the newest member of a family of ITU umbrella recommendations that cover videotelephony and multimedia communications over a variety of pipelines. H.323 is in many ways a derivative of H.320, a 1990 umbrella recommendation for video telephony over switched digital telephone networks. H.323 borrows heavily from H.320's structure, modularity, and audio/video codec recommendations.

Interoperability

In the past few years, interoperability testing has come to the forefront of the conferencing industry. Sponsored by the IMTC and dozens of individual hardware and software companies, interoperability testing enables developers to test their H.32x- and T.120- compliant products with others.

While the ITU's role is that of a standards-setting body, the IMTC focuses on the practical validation and promotion of standards. The IMTC's emphasis is on multimedia tele-conferencing, including still-image graphics, full-motion video, and data teleconferencing. The IMTC focused on ensuring the adoption of the required standards and education of the market.

IMTC-organized events are intended to facilitate the rapid development and delivery of standards-based conferencing products and services and to continue promoting the importance of industry-wide interoperability as a base for building consumer confidence. To date, the interoperability tests have centered on H.324 and T.120 testing. H.323 interoperability tests began in 1996 and will continue through the coming years. Testing is likely to extend over a protracted period of time as multiple vendors cooperate to test a multi-dimensional matrix of equipment, networks, codecs, and protocols.

Implementing H.323

With H.323 standards beginning to take root in the market, equipment vendors and software providers face the challenge of implementing the complex H.323 standard. DataBeam provides developers with a set of software toolkits and development platforms to implement all or a portion of the H.323 standard. These toolkits will enable vendors to select the functionality they need to complement their product and ensure that it will work with other vendor products. DataBeam is committed to interoperability with all other H.323-compliant vendors.

DataBeam has made a significant research and development investment in the development of H.323 technology. The code is field-tested and proven in product implementations.

Continued maintenance and updates to the technology will be available through DataBeam. DataBeam is an active participant in standards bodies and tracks changes to the H.323 specification.

DataBeam's experience and expertise in OEM licensing ensures that customers receive the highest quality service and support. DataBeam is committed to providing H.323 developers comprehensive development solutions for the fastest route to market.

Key H.323 Terms

Call: Point-to-point multimedia communication between two H.323 endpoints.

Call Signaling Channel: Reliable channel used to convey call setup messages following Q.931.

Centralized Multipoint Conference: A call in which all participating terminals communicate in a point-to-point fashion with an MCU.

Common Intermediate Format (CIF): Image format for H.263. Represents 352 pixels/line by 288 lines/image.

Decentralized Multipoint Conference: A conference in which the participating terminals multicast to all other participating terminals without an MCU.

E.164: Address format for ISDN networks. See ITU Recommendation E.164 (1991).

Endpoint: A Terminal, Gateway, or MCU.

Gatekeeper (GK): An H.323 entity that provides address translation, control access, and sometimes bandwidth management to the LAN for H.323 terminals, Gateways, and MCUs.

Gateway (GW): An H.323 entity which provides real-time, two-way communications between H.323 terminals on the LAN and other ITU terminals on a WAN, or to another H.323 Gateway.

H.323 Entity: Any H.323 component, including terminals, Gateways, Gatekeepers, MCs, MPs, and MCUs.

H.245 Logical Channel: A channel carrying information streams between two H.323 endpoints.

IP: Internet Protocol

Local Area Network: A shared or switched medium, peer-to-peer communications network which may include inter-networks composed of LANs connected by bridges or routers.

Multicast: A process of transmitting from one source to many destinations. The actual mechanism may be different for different LAN technologies.

Multipoint Conference: A conference between three or more terminals, which may be on the LAN or on the Circuit Switched Network.

Multipoint Control Unit (MCU): An endpoint on the LAN which enables three or more terminals and Gateways to participate in a multipoint conference. The MCU includes a mandatory Multipoint Controller and optional Multipoint Processors.

Multipoint Controller (MC): An entity which provides for the control of three or more terminals in a multipoint conference.

Multipoint Processor (MP): An entity which provides for the processing of audio, video, and/or data streams in a multipoint conference. The MP provides for the mixing, switching, or other processing of media streams under the control of the MC.

Quality of Service (QoS): Guarantees network bandwidth and availability for applications.

Q.931: Call signaling protocol for setup and termination of calls.

RAS Channel: An unreliable channel used to convey the Registration, Admissions and Status messages and bandwidth changes between two H.323 entities.

Reliable Transmission: Connection-oriented data transmission which guarantees sequenced, error-free, flow-controlled transmission of messages to the receiver.

Resource Reservation Protocol (RSVP): IETF specification. Allows applications to request dedicated bandwidth.

Real-Time Protocol/Real-Time Control Protocol (RTP/RTCP): IETF specification for audio and video signal management. Allows applications to synchronize and spoil audio and video information.

Switched Circuit Network (SCN): A public or private switched telecommunications network such as GSTN or ISDN.

TCP: Transmission control protocol. A reliable networking layer on top of IP.

Terminal: An endpoint which provides for real-time, two-way communications with another Terminal, Gateway, or MCU. A terminal must provide audio and may also provide video and/or data.

UDP: User Datagram Protocol. An unreliable networking layer which sits at the same level of the networking stack as TCP.

Unreliable Transmission: Connection-less transmission which provides best-effort delivery of data packets. Messages transmitted by the sender may be lost, duplicated, or received out of sequence.

Zone: A collection of all Terminals, Gateways, and MCUs managed by a single Gatekeeper. A zone must include at least one Terminal and may include LAN segments connected using routers.

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